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Applied Research Laboratory

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Technical Report

USE OF PROXIMITY EFFECT IN HEARING
AID MICROPHONES TO INCREASE TELEPHONE
INTELLIGIBILITY IN NOISE

by

Kathryn R. Wilson

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The Pennsylvania State University
APPLIED RESEARCH LABORATORY
P. O. Box 30
State College, PA   16804

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Technical Report No. TR 88-007

July 1988

Unclassified
SECURITY CLASSIFICATION OF THIS PAGE

REPORT DOCUMENTATION PAGE

1a. REPORT SECURITY CLASSIFICATION Unclassified		1b. RESTRICTIVE MARKINGS			
2a. SECURITY CLASSIFICATION AUTHORITY		3 DISTRIBUTION/AVAILABILITY OF REPORT Unlimited			
2b. DECLASSIFICATION/DOWNGRADING SCHEDULE					
4 PERFORMING ORGANIZATION REPORT NUMBER(S)		5. MONITORING ORGANIZATION REPORT NUMBER(S)			
6a NAME OF PERFORMING ORGANIZATION Applied Research Laboratory The Pennsylvania State University	6b OFFICE SYMBOL (If applicable) ARL	7a. NAME OF MONITORING ORGANIZATION Naval Sea Systems Command Department of the Navy			
6c ADDRESS (City, State, and ZIP Code) P. O. Box 30 State College, PA 16804		7b. ADDRESS (City, State, and ZIP Code) Washington, DC 20362			
8a. NAME OF FUNDING/SPONSORING ORGANIZATION Naval Sea Systems Command	8b. OFFICE SYMBOL (If applicable) NAVSEA	9. PROCUREMENT INSTRUMENT IDENTIFICATION NUMBER			
8c. ADDRESS (City, State, and ZIP Code) Department of the Navy Washington, DC 20362		10. SOURCE OF FUNDING NUMBERS			
		PROGRAM ELEMENT NO.	PROJECT NO.	TASK NO.	WORK UNIT ACCESSION NO.

11 TITLE (Include Security Classification)
Use of Proximity Effect in Hearing Aid Microphones to
Increase Telephone Intelligibility in Noise

12 PERSONAL AUTHOR(S)
Kathryn R. Wilson

13a TYPE OF REPORT M.S. Thesis	13b. TIME COVERED FROM _____ TO _____	14. DATE OF REPORT (Year, Month, Day) July 1988	15. PAGE COUNT 82

16 SUPPLEMENTARY NOTATION

17	COSATI CODES		18. SUBJECT TERMS (Continue on reverse if necessary and identify by block number)
FIELD	GROUP	SUB-GROUP	electroacoustic systems, hearing aids, microphones, proximity effect, signal intelligibility, signal interference, telephone equipment, telephone interference and signal

19 ABSTRACT (Continue on reverse if necessary and identify by block number)

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proximity effect to increase the intelligibility of telephone speech

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20 DISTRIBUTION/AVAILABILITY OF ABSTRACT ☒ UNCLASSIFIED/UNLIMITED ☐ SAME AS RPT. ☐ DTIC USERS	21. ABSTRACT SECURITY CLASSIFICATION Unclassified	
22a NAME OF RESPONSIBLE INDIVIDUAL	22b. TELEPHONE (Include Area Code)	22c. OFFICE SYMBOL

DD FORM 1473, 84 MAR   83 APR edition may be used until exhausted.
All other editions are obsolete.

SECURITY CLASSIFICATION OF THIS PAGE
Unclassified

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pressure-gradient microphone (bi-directional), a zero-order

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(cardioid). The signal picked up by these microphones was recorded

and played back to normal-hearing subjects through a modified hearing

aid, while the Multi-Talker noise was played in the background. The

pressure gradient microphones allowed significantly better

understanding of the telephone signal than did the pressure

microphone and this difference was more pronounced at higher noise

levels. The bidirectional and cardioid microphones did not provide

significantly different scores at any noise level. It is argued that

this similarity may be due to head effects reducing the pressure-

gradient sensitivity of the microphones. The use of the proximity

effect to enable hearing aids to pick up a telephone conversation

while discriminating against background noise appears to be

successful.

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ABSTRACT

This thesis describes an experiment to test the use of the proximity effect to increase the intelligibility of telephone speech for hearing-aid wearers. NU-6 word lists were played through the equivalent of long-distance telephone lines with a standard Bell 500 handset, while Multi-Talker noise was played in the background at three different levels. The signals were picked up with one of three microphones placed by the ear of a dummy head: a first-order pressure-gradient microphone (bi-directional), a zero-order microphone (omni-directional), or one with order between zero and one (cardioid). The signal picked up by these microphones was recorded and played back to normal-hearing subjects through a modified hearing aid, while the Multi-Talker noise was played in the background. The pressure gradient microphones allowed significantly better understanding of the telephone signal than did the pressure microphone and this difference was more pronounced at higher noise levels. The bidirectional and cardioid microphones did not provide significantly different scores at any noise level. It is argued that this similarity may be due to head effects reducing the pressure-gradient sensitivity of the microphones. The use of the proximity effect to enable hearing aids to pick up a telephone conversation while discriminating against background noise appears to be successful.

TABLE OF CONTENTS

TABLE OF CONTENTS (Continued)

LIST OF TABLES

LIST OF FIGURES

ACKNOWLEDGEMENTS

CHAPTER 1

INTRODUCTION

1.1 General

The telephone has become an increasingly vital part of the lives
of modern people. Telephone technology has increased throughout the
century, making the telephone an integral part of daily life for
large numbers of people. Unfortunately, a significant minority,
those with a hearing impairment, are not able to have full access to
telephones. Some of the technological advances, which were designed
to improve communications for everyone, have hampered communication
for those who wear hearing aids.

Everyone, whether or not he wears a hearing aid, will understand
more of telephone speech in a quiet room than in a noisy one (Holmes,
Frank, and Stoker, 1983). This is because the background noise,
transmitted to the ear by two different paths, masks the desired
signal. One of these paths is direct transmission to the ear (or to
the hearing aid), due to poor coupling of the handset to the ear.
The second path is the side-tone feedback. In normal conversation,
this feedback from the microphone to the earphone of the handset
allows users to monitor their voice levels. At high background noise
levels, this feedback path will permit increased noise that
contaminates the telephone speech and reduces telephone listening
ability (Holmes, Frank, and Stoker, 1983).

Hearing aid wearers have even more problems with the background noise, since hearing aids usually do not couple well with the telephone handset (Lybarger, 1982).

In the past, hearing aids have been designed to reduce background noise contamination by using a non-acoustic method of coupling. Instead of using the microphone to pick up the acoustic output of the telephone, the hearing aid could be switched to the telephone setting and an induction coil would pick up the electro-magnetic leakage from the telephone. This method was first implemented in 1936 and soon after became a standard feature of many hearing aids. Unfortunately for the hearing impaired, leaky telephones are very inefficient, and ensuring that every telephone leak sufficient amounts for a hearing aid telecoil would cost telephone companies millions of dollars per year (Lybarger, 1980). This cost would be shared among all telephone users, for a benefit to a very small percentage.

The replacement of leaky phones with ones that are efficient has meant that another method of coupling telephones with hearing aids needs to be developed. In order that the method work with all telephones, it must operate entirely off the acoustic output of the telephone. This thesis discusses one possible method for improving the acoustic coupling between hearing aids and modern telephones.

1.2 Statement of Problem

The importance of telephone communications in modern life creates the need to make all telephones intelligible for every person, including the hearing impaired. At the present time, there are several options for the hearing impaired. One is the magnetic pickup already mentioned, with the electromagnetic field provided by either a leaky handset or a portable acoustic-to-magnetic converter (Castle, 1981). Another option is amplifying the acoustic output with a specially designed handset, or with a portable amplifier. Other options also involve the use of the acoustic output of the telephone (Lybarger, 1980). Ideally, telephones should be usable by both hearing impaired and normally hearing people, however, this is not the case for many telephones at the present time.

For those people with a mild hearing loss who do not wear aids as well as for hearing aid wearers with gradually sloping or moderate flat hearing losses, amplification of the acoustic signal may be the best solution (Holmes, 1982; Castle, 1981), but it limits their telecommunication to telephones with the special handset or requires a portable amplifier. For those with a severe or total loss, telecommunication devices for the deaf (TDD) are the only solution other than recruiting a normally hearing person to sign or lip speak. In order to allow the hearing impaired to use all telephones, the aid must be able to pick up the acoustic output of the telephone. However, it is a problem to provide a method of coupling the hearing

aid to the telephone handset so that the frequency response is adequate, and the minimal competing noise is picked up. (Stoker, Craig, French-St. George, and Holmes, 1982; Stoker, 1981; Stoker, French-St. George, and Janota, 1983.)

The primary solution in the past has been to switch from acoustic to electro-magnetic coupling. This is done by placing a telecoil in the hearing aid, which can be switched on instead of the microphone. The telecoil picks up the electro-magnetic signal leaked from the telephone receiver and sends it through the aid to the ear. This method of coupling eliminates all room noise except that picked up through the sidetone. A telecoil has been a standard component of most hearing aids since the 1940s, although it is not without drawbacks (Lybarger, 1982). One is that the gain of the telecoil may be different from that of the microphone (Castle, 1981). The second, and more important problem, is that progress in transducer technology has nearly eliminated the magnetic leakage from the receivers, at the same time reducing manufacturing costs and the power required (Lybarger, 1982). Although the hearing impaired can choose an older-type phone for their home or office, a quickly growing minority of telephones, including some pay phones, are not compatible with telecoils.

Another common solution to the problem is to use an amplified handset (Castle, 1981). This method has the advantage of not changing the frequency response significantly. However, it does

require a special handset, which limits its use to special phones, usually only in the office or home.

One approach to this problem of insufficient electro-magnetic signal is to use a device that converts the acoustical output of the telephone to an electro-magnetic signal which is picked up by the telecoil. This allows the hearing aid to be set to "telecoil" with any telephone handset. A similar solution is a device that amplifies the telephone signal, turning any handset into an amplified one. However, these devices are bulky and inconvenient, and more importantly, have very poor coupling with non-standard shaped handsets (Castle, 1981).

In order that the hearing impaired be able to use all phones, some method of acoustic coupling must be used. For in-the-ear (ITE) aids that are not vented, the phone can operate into a sealed volume, reducing the chance of picking up room noise. If an ITE aid is vented, however, there is a serious problem with feedback (Lybarger, 1982). Arndt and Wojcik (1977), propose a frontal tube for behind-the-ear (BTE) aids. This allows a good seal along the cheek and eliminates the feedback problem, but is cosmetically unappealing.

There is ongoing research in the use of adaptive filtering to prevent the background noise from masking the telephone signal for hearing aid wearers (Graupe, Causey, and Soffa, 1977; Cole, 1977). This option is too bulky and expensive to be practical, and the situation is unlikely to change in the near future (Stoker, 1982a).

Some method of increasing the acoustic coupling between hearing
aids and telephones needs to be developed for the hearing impaired to
fully understand telephone speech. The coupling should not introduce
frequency distortion, but should discriminate against competing room
noise. Frequency distortion can be caused by improper acoustic
termination of the receiver, or by the use of a transducer (either a
microphone or a magnetic pickup) that has a very different frequency
response from the normal microphone. Unwanted noise reaches the
listener's ear through the opposing ear, the sidetone, and any air
gaps between the ear (or microphone) and the earphone. All telephone
listeners have to contend with some frequency selective transfer
functions and background noise, but the hearing impaired have even
more difficulty. This thesis studies a hearing aid microphone that
makes use of the spherical wave proximity effect to meet these
requirements.

1.3 Research Objectives

The primary objective of the research presented in this thesis
is to determine if the use of the proximity effect has any benefits
for normally hearing individuals' ability to understand telephone
speech in noise environments.

The proximity effect refers to the increase in sensitivity to
nearfield sources for a microphone that is sensitive to the pressure
gradient. The proximity effect can be used to increase the signal-

to-noise ratio of the microphone when the desired signal is nearby and the noise source is farther away (Olson, 1946; Olson, 1957; Gayford, 1961; Sessler and West, 1969). The physical explanation is that, for a spherical source, the pressure gradient varies as one over the _square_ of the distance from the source, whereas the pressure varies as merely one over the distance.

The three microphones used in this experiment have various levels of sensitivity to the pressure gradient. The omnidirectional microphone had no sensitivity to the pressure gradient (order zero), the bidirectional microphone responded exactly to the gradient (order one), while the cardioid microphone was intermediate between the others (Gayford, 1971; Olson, 1946, 1957). Higher order gradient microphones are available (Olson, 1957), but are not used in this study. The derivation of the increase in sensitivity as a function of frequency (f) and distance from the source (r) for gradient microphones of different orders are in Appendix A. For this experiment the microphone is 5 to 20 dB more sensitive to the telephone than to the noise for the frequency range of interest (see Appendix A).

Since a telephone is nearly always closer to the ear than any competing noise, the proximity effect seems to be the ideal tool for hearing aids to increase the intelligibility of telephone speech. It can provide noise discrimination where other noise reduction methods

may have difficulties, such as with noise that is transient or of the same frequency range as the desired goal.

1.4 General Outline of Thesis

Chapter 1 provides the background of the problem of telecommunication for the hearing impaired: the various factors involved in studying the problem, its solution in the past, and the proposed solution. A general description of previous research in this area in given in Chapter 2. This chapter is broken up into sections that describe studies of electro-acoustical systems and the human systems involved in telecommunication.

Chapter 3 details the rationale behind the experimental methods used in this thesis. Section 3.2 describes the choice of speech signal, which were NU-6 monosyllabic word lists; and the background signal, which was Multi-Talker Noise. The selection and training of the subjects is reported in section 3.3.

The test apparatus used for the original signals, recording and playback are described in section 3.4. This section includes the justification of recording the signals with a dummy head instead of testing in a live situation. The experimental design, which was based on the use of two nine-by-nine Graeco-Latin squares, is outlined in section 3.5.

Chapter 4 contains the results of the experiment and the analysis of these results. The pilot study is detailed in section

4.2, and the results of the main study are in section 4.3. The final section 4.4, contains an analysis of the results, and the conclusions that can be drawn.

CHAPTER 2

REVIEW OF TELEPHONE, HEARING AID, AND INTELLIGIBILITY STUDIES

2.1 Electro-acoustical Systems

Electro-acoustical systems are, in general, a network of transducers and circuits that receive a signal in one medium and convert it to a signal in the other. In some systems, this cycle is then reversed, to reconstruct the original signal. There are systems that convert from electrical to acoustical, but many convert from acoustical to electrical and back to acoustical. Examples are radio (acoustical to electrical to electromagnetical to electrical and back to acoustical), television, public address systems, hearing aids and telephones.

This thesis deals principally with hearing aids and telephones. Both telephones and hearing aids receive the incoming acoustical signal with a microphone. This signal is then sent through an electrical circuit that may reshape the frequency response or amplify the signal. Some of these modifications are intentional and designed to increase intelligibility, while others are unintentional, and degrade speech quality. The signal is then sent through another transducer (earphone or speaker), which outputs an acoustical signal. Both telephones and hearing aids are designed for the acoustical output to be sent to human ears. This means that they should,

ideally, optimize the signal for the recipient's hearing. This usually means that the frequency content of the signal not be radically changed or distorted, although sometimes a less radical change may be desirable. Regardless of the type of telephone, it will pick up background noise as well as the speaker, since the speaker's mouth is only loosely coupled to the microphone, leaving an air gap for the noise to pass through.

The standard telephone handset includes feedback, called the sidetone, which is provided to allow talkers to monitor their voice levels. The amplitude of the sidetone will vary from manufacturer to manufacturer. Unfortunately, if there is any background noise in the listener's room, it will be picked up by the microphone, added to the transmitted signal, and sent to the listener's ear, with a resultant degradation in speech quality. If the background noise is loud enough, it may mask the transmitted signal. Holmes, Frank and Stoker (1983), found that the masking could be significantly reduced by eliminating the sidetone electronically or by covering the microphone with one hand. Unfortunately, covering and uncovering the microphone with one's hand becomes very tedious and thus undesirable.

The electrical network between two telephones adds noise and distortion to the signal, although improvements in transmission quality are always being made. In North America, the network between the two telephones is often very complicated, involving switching at both the local and the long-distance level. The switching may be

either analog or digital, resulting in the possibility of the signal undergoing several conversions from analog to digital and vice versa (Gruber, Bhullar & Williams, 1982). This means that both background analog noise and speech-correlated digital noise are added to the speech signal, lowering the transmission quality.

Frequency distortion of the signal is caused in part by deficiencies in the handsets and the connections between the sets and the switching office. Another cause of distortion is reverberation or listener echo. This echo is caused by imperfect return losses at the connection between the local analog to digital converter and the central host switch. For talker echo, there is one reflection, at the listener's end. The talker hears an echo of her own voice, which may disturb her speech. With listener echo, two reflections occur. This causes the listener to hear both the original signal and a delayed signal. This can have drastic consequences on the intelligibility, depending on the time delay and the amplitude of the echo (Gruber, Bhullar, and Williams, 1982). Elliot (1982) found that reverberation and noise have more impact on the speech perception of the hearing impaired than those with normal hearing.

Hearing aids, like telephones, receive an acoustical signal with a microphone that converts it to an electrical signal, which is then sent to an electrical network, back to acoustical, and then to the ear. However, hearing aids are designed to modify the electrical

signal by amplifying it, and by changing the frequency characteristics to compensate for the loss of hearing.

For those who wear hearing aids, there are noise, distortion and other complications when trying to hear a conversation. The hearing aid itself will add electronic noise, leading to a modified frequency response (not necessarily for the better), and other distortion. Much work has been done in this area; a good source of basic information is the work done by Staab (1978), and Studebaker and Bess (1982). Regrettably, while there has been a great deal of research into the problems and solutions of hearing aids, little investigation of the special problems posed by the use of hearing aids with telephones has been done. What research has been done has mostly been with the support of Bell Northern Research Laboratories (Stoker, 1982a and 1982b; Wojcik and Arndt, 1977; Arndt and Wojcik, 1977).

Castle (1981) and Lybarger (1982) describe the various methods currently used in telephone communication by the hearing impaired. For those who do not wear hearing aids, amplification is the only currently available solution. Amplification also benefits some who do wear hearing aids (Hutchinson, 1983). Amplified handsets can increase the level of the signal by as much as 30 dB, which increases the signal-to-noise ratio. These handsets are very durable, and seem like a good solution to the problem. However, amplified handsets have to be bought separately, and are only a solution when the person is at

that specific telephone. The majority of telephones are not amplified, and thus will not provide acceptable speech levels for the hearing impaired.

One solution that works with most telephones is the use of portable amplifiers. These are relatively small and battery operated. Most use magnetic pickup, but some are now using acoustic pickup. Unfortunately, portable amplifiers are bulky and often do not couple well with handsets due to non-standard shaped receivers. When those with acoustic pickup are used with hearing aids, there is an increased potential for noise because the signal is converted from acoustical to electrical and back three times before finally impinging on the ear.

2.2 Speech Systems

Although speech is a common word and one which is generally understood, the process itself is highly complicated. In normal face-to-face conversations, the speaker creates a series of acoustical signals that are transmitted to the listener's ear. The ear converts these acoustical signals into electrical impulses sent through the auditory nerves to the brain. It is the brain that interprets the meaning of these physical signals. In face-to-face communication, information is also sent by visual means. This includes speechreading, which reinforces the acoustic message, as well as non-verbal cues.

Research into speech and its perception has been continuing for many years, but has not yet provided all the answers. There are many variables in speech perception studies, such as: the listener's familiarity with the words, the number of speech sounds in the words, and the listener's level of training (Moser and Dreher, 1955); the effect of the talker (Resnick, 1962); the age of the listener (Elliot, 1982); and the type of hearing loss of the listener (Suter, 1985). An example of the difficulty in quantifying speech perception is the study by Stoker, Craig, French-St. George, and Holmes (1981), which compared four standard speech discrimination tests and discovered that there was a wide variation in their ability to differentiate between types of subjects and between test conditions. The frequency and temporal content of the speech signal is modified by the talker's physiology, emotions, and also by the language that the talker uses. Jakobson, Fant, and Halle (1967) discuss the distinctive temporal and frequency features of some sixty different languages.

After the speech signal leaves the talker's mouth, it is transmitted through some medium to the listener. The most simple and most common medium is air. Even this has added complications such as wind, competing noise, reverberation, and large distance between talker and listener. There are numerous paths that are much more complex than face to face, with just air between the talker and listener. These involve some sort of transducer, usually converting

the acoustical signal to an electrical signal, sometimes to a mechanical signal. After passing through various devices that may reshape the signal, and perhaps after being recorded in some manner, the signal is eventually converted back to an acoustical signal. For ideal transmission, this final acoustical signal should contain all the information contained in the input signal.

Finally, the speech reaches the listener's ear. The mechanical components of the listener's ear are the first to have an effect. The ear acts as a transducer, converting the acoustical signal to a series of neural impulses which are sent to the brain. If there is damage to any part of the ear, it will affect the transmission of the message, causing the signal to be distorted and/or attenuated, depending on the type of damage. When the sequence of neural impulses reaches the brain, it must be sorted and interpreted if the listener is to understand the message contained in the speech.

Human hearing has a sensitivity and range almost unmatched, even in the most elaborate, high-fidelity apparatus. The ear can pick up signals that are so small that the displacement of the eardrum is less than 1% of the diameter or a hydrogen molecule (Stevens and Davis, 1983, p. 56). Stevens and Davis also conclude that someone with acute hearing is able to detect changes in pressure of the order of magnitude as that due to brownian motion (thermal noise). The upper limit of hearing is the threshold of pain. This is as much as 120 dB higher than the threshold of hearing for normal listeners

(ibid). This dynamic range is rarely duplicated in an electro-mechanical system.

The ear and brain form an extremely complex signal processor. A person can detect minute changes in sound (Fletcher, 1929), and yet is able to allow for a great deal of flexibility in the exact frequency and temporal content of speech. The brain can recognize a familiar voice, even if the words are not familiar or the talker's voice is changed due to excitement or a cold. It can also understand the meaning of speech produced by a person whom we have never heard before. Humans may even be able to understand an unfamiliar word, if it is heard in context, since speech signals contain much duplicate information. In computerized voice recognition devices, at least three problems have yet to be fully solved. The first is recognition of speech that is insensitive to variations in the voice of the talker, such as those due to a cold or excitement. The second is recognition of speech by different talkers. The last difficulty is the identification of a word that the machine has not heard before. In short, the mental processes used to recognize and understand speech are very complex and are not completely understood.

Similar problems occur with the representation of speech for testing. Just as artificial intelligence can only implement very basic speech recognition, so speech perception testing can only use very simplified speech signals. When speech is broken into its

fundamental components so that it can be reliably tested, the results may not reflect actual speech perception conditions.

Studies of speech communication systems must use some signal to represent speech. They must also use either human subjects or some model of speech intelligibility. The different signals and testing procedures are chosen to be natural and reliable. Unfortunately, naturalism and reliability are quite often mutually exclusive. This means that the researcher must strike a balance between these two criteria. In general, the more simple the signal, the more repeatable, and the more natural the signal, the more complex and the less reliable. Signals that have been used range from discrete tones to continuous discourse (Moser and Duker, 1955; Hirsh, 1952). The type of signal chosen also depends on the precise aspect of hearing that is being investigated (Stoker, Craig, French-St. George and Holmes, 1981).

The most simple signal that is commonly used is pure tones. This signal allows for fast testing, high repeatability, and easy comparison between different experiments (even those done in different laboratories with different equipment). The most commonly used frequencies are 500, 1000, 2000, 3000, 4000, 6000, and 8000 Hertz, because these frequencies are representative of the frequency content of speech (Vasallo and Glorig, 1975). This means that the subject's sensitivity to these tones gives a rough estimate of his or her ability to understand speech. Elliot (1982) showed that the age

of the subjects had a marked effect on the measured speech

intelligibility, even though all her subjects had normal hearing.

The next level of naturalism is to use nonsense syllables. In

this method, the English (or other) language is divided into its

component parts or phonemes. This means that the signal has much the

same frequency and temporal distribution as normal speech, without

adding the further complexity of meaning. The difficulty with

nonsense syllables is in recording what the subject hears. Either

the subjects must be familiar with phonetic transcription, or they

must repeat what they hear. Limiting subjects to those that are

familiar with phonetics either drastically limits the pool of

available subjects, or requires intensive and extensive training of

naive listeners so that they are conversant with phonetic

transcriptions. If oral repetition is used, the hearing ability and

bias of the person recording the results is also tested, because the

subjects repeat the word that they hear, and the recorder writes it

down. In many cases the researchers are the ones who record the

subjects' responses and consequently the researchers may hear what

they want to be hearing, rather than what the subjects are actually

saying.

In order to eliminate the need for phonetic transcriptions, and

increase the naturalism, monosyllabic words from the English language

are used. This method originated in the Bell Telephone Labs with the

work of Fletcher and his colleagues (Fletcher, 1929 and 1953;

Fletcher and Galt, 1950), and gained common usage with the development of the Harvard Psycho-Acoustic Laboratory's Phonetically Balanced (PAL-PB) lists of monosyllabic words. The PAL-PB-50 was designed to contain 20 lists of 50 words, with no duplication (Tillman et al. 1963; Resnick, 1962).

Resnick details the criteria used in designing the PAL-PB-50 lists: "monosyllabic structure, equal average difficulty, equal average range of difficulty, equal phonetic composition, composition representative of English speech, and words in common usage." The test-retest reliability for the PAL-PB-50 lists is quite high. A disadvantage of these lists is that some of the words are not familiar to subjects with a limited vocabulary, which affects their performance (Tillman et al. 1963). Several other monosyllabic tests have been developed using the same phonetic balance, such as the PB-K-50 (Haskins, 1949), and the W-22 (Hirsh et al., 1952). Others have revised the criteria for a desired phonetic balance, and designed new lists accordingly. The most commonly used of these tests are the Fairbanks Rhyme test (Fairbanks, 1958), the Lehiste-Peterson Consonant-Nucleus-Consonant (CNC) lists (Lehiste and Peterson, 1959; Peterson and Lehiste, 1962), and the Northwestern University tests NU-4 (Tillman, Carhart and Wilber, 1963) and NU-6 (Tillman and Carhart, 1966).

Although all monosyllabic word tests have similarities, they each distinguish slightly different attributes of speech. This means

that there is no "best" test for all applications. Different researchers developed new tests to meet some criterion of repeatability or representation of the English language not adequately met in previous tests.

A disadvantage of the W-22 lists, for example, is the inability to differentiate sharply among small variations in phonemic discrimination (Tillman, Carhart and Wilber, 1963). The Fairbanks Rhyme test can distinguish these minor variations, but solely for consonants. It also has the drawbacks of having a very limited pool of words and no definitive relationship to the more standard PB lists.

The Lehiste-Peterson CNC lists have a phonemic balance that is matched to that of English monosyllabic words. This balance is somewhat different from that of all English words. The words in these lists were chosen out of the most familiar CNC words in the English language, so that performance will not be affected by limited vocabulary. During the development of these lists, careful analysis of "phonetic, phonemic, and linguistic considerations" was conducted prior to choosing only CNC words rather than any monosyllabic words (Tillman et al., 1962; Lehiste and Peterson, 1959; Peterson and Lehiste, 1962).

The Lehiste-Peterson CNC lists were further modified by researchers at Northwestern University. This research led to the

development of the NU-4 and then the NU-6 lists. These lists not only have a more ideal phonemic balance than the Lehiste-Peterson CNC lists, but they also have had the reliability and interchangeability of the lists thoroughly checked (Tillman et al., 1963; Tillman and Carhart, 1966).

The choice of list used in an experiment depends on the exact type of evaluation desired. Stoker, Craig, French-St. George, and Holmes (1981) conducted a comparison between four standard speech discrimination tests for the specific application of evaluating telephone-hearing aid coupling. They found that none of the tests were "best" for all listening conditions, or for all types of evaluation. This means that the researcher must decide which measures of speech discrimination are the most appropriate. Some of the measures to choose from are intra-subject differences, differences between certain groups of subjects (and which groups are to be differentiated), and phonemic error analysis or language content.

The choice of the content of the speech discrimination test is only the first complication in designing an experiment. Many other variables also have an effect, so that very few experiments can be directly compared. The manner of presenting the material sometimes has a greater effect on discrimination scores than the words that are in the test. Resnick (1962) found that there is greater test-retest reliability using the same talker and different lists than using the

same list and different talkers. This is corroborated by the work of Frank and Craig (1984) which compares word discrimination scores using two different recordings of the NU-6 lists in quiet and with background noise. Their results show an increasing difference between the discrimination scores as the signal-to-noise ratio decreased. They suggest several factors which may cause the difference. One is that the two recordings use different carrier phrases. Another is that different talkers were used. Their conclusion, however, is that the discrepancy in the scores is principally due to errors in the recording procedure when taping the lists, which resulted in one recording having its test word level 6 dB higher than the other, invalidating the chosen signal-to-noise ratio.

Although many of the word lists have commercially available recordings, live voice readings of the lists are quite common. This adds the talkers' voices, as well as their ability to monitor their speech, to the list of variables. It requires training to be able to read each carrier phrase and word with constant tempo, clarity and (most importantly) level. There are also the variations in the test system. Even when the same equipment is used, the talker may not be a constant distance from the microphone or from the subject's ear. This would affect the signal level, the frequency shaping, and the effect of reverberation, since the talker might be in a different

part of the room. Recording the signal adds extra equipment to the speech path which can add noise and distortion. On the other hand, recordings do allow direct comparisons between lists run at different times or by different people or even in different sites, which is not possible with live voice tests.

Because no current intelligibility test is ideal for all testing situations and subjects, and because there is wide variation in the method of presenting test materials, accurate comparisons between different studies is not possible. Until more ideal tests and presentation methods are developed, care must be taken in choosing the best for the specific experiment, and then documenting precisely how the experiment was conducted.

It must be remembered that monosyllabic word intelligibility is only an approximation to everyday speech perception, since it limits the possible pool of words to those of one syllable. Also, in everyday speech, some of the information is redundant, so that part or all of a word may be predicted from the rest of the sentence. Thus, an intelligibility score of only 70%, measured according to American National Standard S3.2-1960 (R1976), will give reliable communication (ANSI S3.14-1977). The American National Standard methods for the calculation of Articulation Index (ANSI S3.5-1969) includes a comparison of the percent test components understood versus Articulation Index for several types of test.

More complicated testing materials are sometimes used to more accurately represent speech, although this adds further complications and lessens test repeatability. For instance, some tests utilize spondees (disyllabic words with equal stress on both syllables) which are much less common than monosyllabic words, and thus are more predictable. Another commonly used set of testing materials is complete sentences. The original intelligibility studies at the Bell Telephone Labs used interrogative sentences (Fletcher, 1929). The problem with this type of test is that it not only tests the subject's ability to hear and understand speech, but his or her knowledge of the answer. The questions used by Fletcher often required familiarity with New York City, limiting the geographical region in which the list was useful.

The most natural (and least analytic) type of test material is continuous discourse. For this test, some uniform sample of English is read, and the subject is given instructions on evaluating the quality of speech reception. Some of the more commonly used measures of speech quality are intelligibility (or clarity) and preferred listening level. The specific instructions given will influence the subject's response, so care must be taken to give exactly the same instructions to all subjects (Moser and Dreher, 1955). There are also different techniques for recording the response. One is to provide the subject with two different conditions, and forcing a choice of which is "better." A similar method is to allow the

subject to vary whatever physical dimension is being studied until it is at the "best" position. Yet another procedure is to have the subjects assign a grade to the signal.

A difficulty with subjective testing in general is that the preferred signal is not necessarily the one with the highest intelligibility. Cunningham, Merle, and Drake (1978) found no correlation between hearing aid wearer's level of satisfaction with an aid and discrimination ability. Lutman and Clark (1986) found a similar result in a comparison of simulated hearing aids: "Speech test performance and subjective preference or rating gave contradictory indications (p. 1039). " This means that some other criteria are being used by the subjects. Until those criteria are discovered, and some method of measuring the criteria is found, speech discrimination is one of the best measures of the success of a hearing aid.

CHAPTER 3

METHOD

3.1 General

This chapter describes the method used to test the hypothesis

that the pressure gradient effect can be used to increase the

intelligibility of telephone speech for hearing aid wearers.

Initially, a quick pilot study was run to provide preliminary

information on the usefulness of continuing the investigation. The

main experiment was conducted over a two month period using three

different microphones for the prototype hearing aid, and with three

levels of competing background noise. The subjects for both studies

were university students or staff with normal hearing. The test was

designed to minimize any effects due to differences of word list,

subject or order of presentation.

Section 3.2 describes the selection of the word list and

competing background noise used in this experiment. The word lists

used were the NU-6 lists copied from master tapes provided by Auditec

of St. Louis. The background noise used was multi-talker noise, also

copied from Auditec master tapes. The multi-talker noise was set at

three different sound pressure levels during the test.

The subject selection rationale and subject preparation for the

tests are described in Section 3.3. Section 3.4 describes the test

apparatus used to generate and record the telephone and background

noise signals, that used to play the signals back to the subjects,

and the method of recording the results. The experimental technique

was designed so that differences between word lists, subjects, and

order of presentation would not bias the data. The theory behind

this design, as well as a description of the design, is described in

Section 3.5.

Section 3.6 provides the details of the analysis methods used

for the experiment. The measure of interest was the percent words

understood correctly with each microphone at each noise level. The

three microphones have different levels of proximity effect and

directionality. The construction of the experiment was such that

extraneous variation due to learning effect, subject and word lists

was minimized, allowing the effects of microphone and noise level to

be clearly distinguished.

3.2 Choice of Speech Signal and Background Noise

The choice of speech signal and of the background noise has a

large effect on the speech intelligibility measured in an experiment.

As described in Section 2.2, the spectrum of repeatability versus

naturalism of test signals used to measure speech reception ranges

from highly repeatable discrete pure tones to the totally natural

complete sentences. Discrete pure tones or bands of noise are highly

repeatable, but only give an approximate measure of the subject's

ability to understand speech. This measure was used to screen the subjects for normal hearing. The next level of realism is nonsense syllables, which contain the same phonemes used in speech. Unfortunately, use of this type of test requires trained subjects. Monosyllabic word lists are often chosen, since they provide a good balance of naturalism and repeatability and were used in this experiment.

The type of speech signal is not the only signal variable in an experiment: there is a choice of competing noise, which has a significant effect on the intelligibility of a speech signal and the best methods to reduce it. In order to test in a worst-case scenario, the competing noise used was Multi-Talker noise (available from Auditec of St. Louis), since it contains the same spectral content as the desired signal. A detailed description of the Multi-Talker noise used is in Appendix B.

3.3 Subject Selection and Preparation

Intelligibility is very difficult to measure accurately, due to the many variables that must be accounted for. The results can be affected by the subjects, by the choice of test procedure, and by variations in recording procedure (Frank and Craig, 1984).

The first variable to be dealt with is the choice of subjects. Since this thesis was designed to assist those hearing impaired who

wear hearing aids, it would seem obvious that the subjects should be hearing aid wearers. There are two major reasons that normal hearing subjects were used instead. The first is that there are a wide variety of hearing losses. Each type of loss has a different effect on the ability to hear speech. Because of this, if the only subject selection criteria was that the person wore a hearing aid, there would be variations in the subject responses due only to the person's specific hearing loss. If a certain type of hearing loss is used as criterion for selection of subjects, the results could show a greater, or a lesser effect than for other hearing aid wearers. For this reason, normal hearing subjects are chosen.

Two series of experiments were run during the course of this investigation. The first was a pilot study which did not include the sidetone or head effects. Subsequently, the more extensive main experiment was run. The following description applies to both experiments.

Ten American male subjects were used for the pilot study, with an average age of 32, ranging from 26 to 44. The main test used 18 subjects, 7 women and 11 men. Their ages ranged from 21 to 34, with an average of 25. All of the subjects were university students or staff. Eight of the pilot study subjects were staff, and two were students. For the main test, however, 16 of the 18 subjects were university students, and only two were staff, which accounts for

the age difference between the two groups. The test was conducted

with the approval of the Office for the Protection of Human

Subjects, with each subject reading and signing an informed consent

form, shown in Appendix B.

The subjects all had hearing thresholds of 20 dB or less at 500,

1000, 2000, 4000, and 8000 Hertz in both ears, as measured on a Maico

audiometer in an IAC audiometry test suite, as described in Vasallo

and Glorig (1975). If the subject met the hearing requirement, he

was then told that a prototype hearing aid would be placed in his

ear, and it would transmit various signals. First would be

continuous discourse with some background noise (to accustom them to

listening), then a man reading nine word lists with background noise

that would be at different levels for each list. The subjects in the

pilot study were told that they should attempt to write down the word

that they heard, and to guess if they were not sure.

Some of the subjects in the pilot study were very concerned that

some of the situations were almost totally unintelligible, and

stopped the test, resulting in the subject missing some of the words

in a list. To prevent this loss of words, and the need to throw out

subject data, the subjects for the main test were told that some of

the situations were so noisy that it would be very difficult to

understand any of the words but to keep trying, and that the next

list would probably be more intelligible.

For both the pilot and main tests, the first signal played through the telephone network was a one-minute segment of continuous discourse to allow them time to adjust to the noise and prepare for the word lists. Then the word lists, with their respective microphone and multi-talker-noise level, were played in their right ears, and they wrote down the words that they heard on the data collection forms provided.

3.4 Test Apparatus

The pilot study was not intended to duplicate the actual conditions of a hearing aid wearer listening to telephone speech, but to determine whether further study on the proximity effect was indicated. A major simplification was attaching the microphone to the telephone handset with a fiberboard ring, not in a position similar to that of a hearing aid microphone. This was done so that all three microphones would be easily interchangable during the test, and to ensure uniformity of position between subjects. Another simplification was that the sidetone was electronically eliminated.

The tests were conducted in a sound-proof room with various noise levels. The noise was multi-talker babble, played from a Crown 700 series reel-to-reel tape player through a Superscope S-2 speaker in a corner of the room. The noise level was calibrated to 45, 55, or 65 dBA at the microphones. The speaker was placed 1 m. from the microphones, with the plane of its face perpendicular to those of the

microphones and telephone receiver. The telephone was placed with the plane of the receiver 12 from the vertical (IEEE, 1983). Tape recordings of Auditec NU-6 half word lists were played from a Sony Cassette-Recorder TC-95L. The lists then went through an APREL mark IV Audiometer Telephone Interface (ATI) which simulates a long distance line (Stoker, 1982). Each tape included a calibration tone so that the voltage could be adjusted to provide 86 dBA at the receiver (ANSI, 1973).

The microphones used were Countryman Associated Isomax II B, II C, and II O, which are nominally bi-directional (first-order pressure gradient), cardioid (order between 0 and 1) and omni-directional (pressure). They were mounted on the handset by a fiberboard ring. They were placed with their major axis of symmetry facing the receiver. The signal picked up by the microphone was sent through a custom designed variable gain amplifier to a modified hearing aid. This aid is a Dyn-Aura model 630 with everything except the shell, tubing and 1606 receiver removed. Individual, disposable earmolds (All-American Mold Co's EAR plug molds) were used. The voltage to the aid was calibrated prior to each test to provide 86 dBA to the ear (ANSI, 1973).

The results of the pilot study showed a marked increase in telephone intelligibility when a microphone sensitive to pressure gradient was used, instead of the usual pressure-sensitive microphone. Because the results of the pilot study were so

encouraging, the main experiment was conducted to see if the hypothesis was still valid with a more realistic setup.

The intention of the main experiment was to provide a preliminary test of the hypothesis that the proximity effect would increase the word discrimination of telephone conversations in relation to ordinary microphones. Because this test was only the first step in assessing the use of the proximity effect in hearing aids, emphasis was placed on repeatability and controlled conditions.

To make the test more like that of typical hearing aid usage, the microphones were placed in a position similar to the microphone on behind-the-ear hearing aids. The microphones were oriented so as to optimize the signal-to-noise ratio (i.e., to maximize the proximity effect). This optimization was carried out by sending a pure tone (1000 Hz) through the telephone handset at 86 dB while playing white noise through the loudspeaker used for the test at 65 dBA at the ear and analyzing the microphone spectra to determine SNR. The bidirectional and omnidirectional microphones were tested in two positions: behind the ear and above the ear of a dummy head. The cardioid was not tested, on the assumption that it's optimal position would be similar to that of the bidirectional microphone.

The telephone was positioned next to the dummy head according to IEEE Std. 269-1983. The microphones were placed (one at a time) in one of the two basic positions, and then the SNR was recorded as the microphone was rotated in increments of 15 about all three of its

axes. Surprisingly, both the bidirectional and omnidirectional microphones had the highest signal-to-noise ratio at the same orientation and position: behind the ear, 75 from the vertical, 45 off forward, and 45 rotation about the microphone axis. This optimal position was tested on a volunteer (another Penn State student) in order to see how well the data from the styrofoam head matched with a live subject, and similar results were found. This position was then used in recording the signals for the main portion of the test.

The dummy head used to simulate a human head was a simple shape formed of styrofoam, with cutouts to allow the insertion of latex ears. Use of a dummy head results in some errors, because the head will not replicate all the acoustical features of actual hearing aid wearers' heads. However, some practical difficulties, and potential problems with repeatability led the researcher to choose the dummy head instead of the subjects' heads.

One of the problems that led to the choice of a dummy head is the physical positioning of the microphones behind the ear. Since the experimental design required the microphones to be tested in alternate order, each subject would have to sit through several changes of microphone during their portion of the test. It took several minutes to replace one microphone with the other, which would have nearly doubled the time required.

A different complication of microphone positioning with live subjects was the exact placement of the microphone and telephone. Some studies on various methods of improving the word discrimination of telephone speech allowed the subjects to hold the telephone as they do in everyday use (Tan et al., 1884; Holmes et al., 1983). This method has the virtue of being highly realistic, since it is the normal, everyday position. Unfortunately, determining and recording the exact position of the telephone is difficult even if the subjects do not move during the test. Since most subjects will move during the test, recording the position of the phone is very difficult, and it is impossible to repeat the test exactly.

This experiment was designed to be the first in a sequence of tests, and as such had a greater emphasis on repeatability than on naturalism. The use of a dummy head eliminated both the need to find some reference axis on each subjects' head, and the variation in signal due to the motion of the handset relative to the subject. A schematic of the experimental arrangement is shown in Figures 3.1 and 3.2.
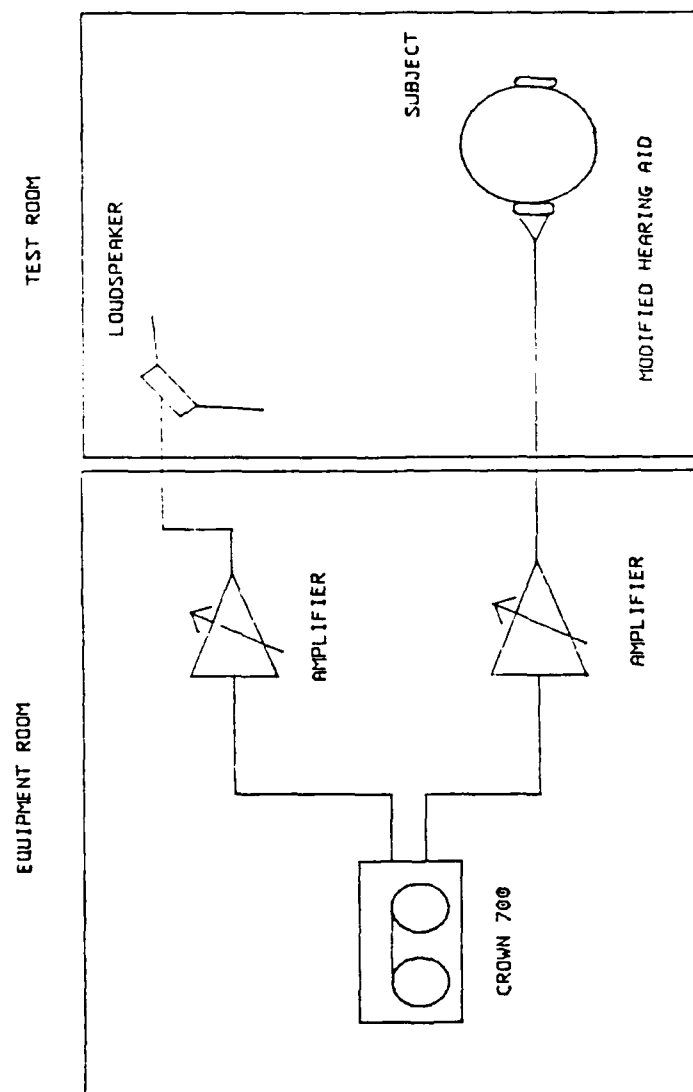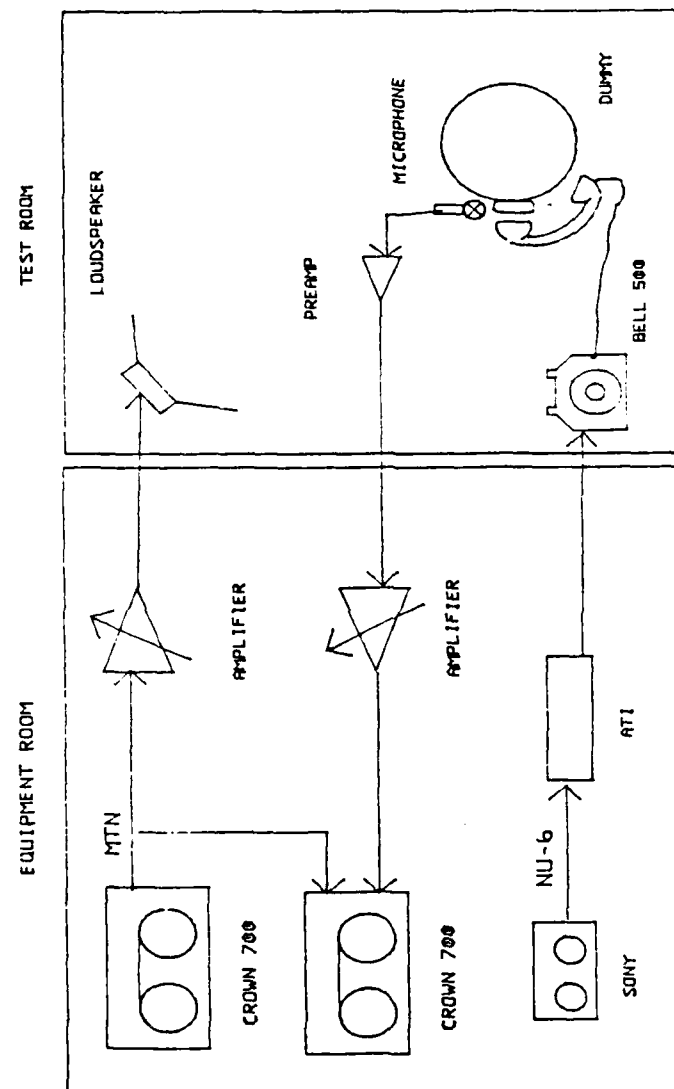
Figure 3.1 Recording Setup

Figure 3.2 Playback Arrangement

The ATI is a device developed by researchers at Bell Northern Research (Stoker, 1982), which was designed to modify the input signal by adding noise and distortion in order to simulate the effects of switching networks and long distance wires. This device provides the worst case test of telephone intelligibility that occurs in normal telephone use. As Stoker (1982) points out, two telephones in the same office have a clearer connection than two across town, and much clearer than a long-distance call.

A test sequence was recorded for each of the subjects, with a total of two sections per sequence. The first section was the continuous discourse with 45 dBA multi-talker noise to the allow the subjects to adjust to the hearing and noise. Following this section were the actual test signals with the multi-talker noise. The nine-word lists, with the appropriate microphone and noise level for each subject were played in the order laid out in the experimental design. During the recording procedure, the word list was calibrated at the handset to a level of 86 dBA (ANSI S3.7-1973 (R1979)), with an artificial ear. The multi-talker noise was calibrated to a level of 45, 55, or 65 dBA at the microphone.

For playback, the subject's tape was played on a Crown 700 series reel-to-reel tape recorder. The multi-talker noise channel was played through a custom designed variable amplifier to the same speaker used for the recording session. The subjects' unoccluded ear was exposed to this noise simultaneously. Since the signal was

recorded directly from the tape, the only phase shift between the subject's ear and the microphone would be due to the small (less than 10 inches) spatial variation in distance from the speaker to the subject and any phase error in the tape recorder itself. The second channel, which carried the microphone output, was sent through a variable gain amplifier to the modified hearing aid and then to the subject's ear. The output of the aid was calibrated to 86 dBA in a 2cc coupler.

The hearing aid used in this portion of the test was the same as that in the pilot study: a Dyn-Aura model 630 with everything except the shell, tubing and 1606 receiver removed. Once again, individual, disposable earmolds (All-American Mold Co's EAR plug molds) were used. The aid was placed in the right ear of each subject.

The subjects wrote down the word that they heard on data collection form. They were instructed to guess if they were unsure, and to try to write legibly. The forms are then scored by marking the words that were written down as correct or incorrect. The scores for each half list, and thus each condition, are recorded as a percentage of words understood. This percentage is used in all subsequent analysis.

3.5 Experimental Design

This investigation was designed to test the hypothesis that the proximity effect can be utilized to increase the intelligibility of

telephone speech for hearing aid wearers. The experiment was laid

out with two main requirements: that it allow for accurate analysis

of the results, and that it minimize any effects other than those

tested in the hypothesis by having a completely balanced design.

Many factors can influence the results of intelligibility

studies. In order to minimize the effects of as many extraneous

factors as possible, the first problem is to determine the nature of

some of the variables that might influence the test. The main

factors that may affect the results were the learning effect,

differences between half word lists, inter-subject differences, and

correlation between any of these factors and the two variables needed

to test the hypothesis: the microphone and noise levels.

After deciding that these were the main factors, the

experimental design was based on two nine-by-nine Graeco-Latin

squares (Cochran and Cox, 1957). The treatments are considered to be

the nine microphone-noise level combinations, which exposes each of

the three microphones to each of the three noise levels exactly once.

The results, in percent words correct, are considered adequately

continuous for analysis of variance (Texter, 1985).

The experimental design is laid out in Tables 3.1 to 3.3. Table

3.1 gives the design in terms of subject, order of presentation, test

condition, and word list. Since 9 subjects seemed minimal for

statistical accuracy, two squares are used in the design. The first

nine subjects' test conditions and word lists were determined from

one square, and the second nine from another. The nomenclature for
the test conditions is given in Table 3.2 for both sets of subjects.
The numbering system used for the word lists is given in Table 3.3.

TABLE 3.1

EXPERIMENTAL DESIGN

Letters Refer to Test Condition (Table 3.2) and Numbers
Refer to the Half Word List Used (Table 3.3).

ORDER OF PRESENTATION

	1	2	3	4	5	6	7	8	9
1	A1	B3	C2	D7	E9	F8	G4	H6	I5
2	B2	C1	A3	E8	F7	D9	H5	I4	G6
3	C3	A2	B1	F9	D8	E7	I6	G5	H4
4	D4	E6	F5	G1	H3	I2	A7	B9	C8
5	E5	F4	D6	H2	I1	G3	B8	C7	A9
6	F6	D5	E4	I3	G2	H1	C9	A8	B7
7	G7	H9	I8	A4	B6	C5	D1	E3	F2
8	H8	I7	G9	B5	C4	A6	E2	F1	D3
9	I9	G8	H7	C3	A5	B4	F3	D2	E1

SUBJECT (rows 1–9)

TABLE 3.2

TEST CONDITION NOMENCLATURE

A: For First 9 Subjects

MICROPHONE

		OMNI	CARDIOID	BI
NOISE	45	A	B	C
LEVEL	55	D	E	F
(dBA)	65	G	H	I

B: For Second 9 Subjects

MICROPHONE

		OMNI	CARDIOID	BI
NOISE	45	E	F	D
LEVEL	55	H	I	G
(dBA)	65	B	C	A

TABLE 3.3

HALF WORD LIST NOMENCLATURE

A: For First 9 Subjects   B: For Second 9 Subjects

Number	Word List		Number	Word List
1	1Aa		1	1Ca
2	1Ab		2	1Cb
3	2Aa		3	2Ca
4	2Ab		4	2Cb
5	3Aa		5	3Ca
6	3Ab		6	3Cb
7	4Aa		7	4Ca
8	4Ab		8.	4Cb
9	3Bb		9	3Da

The symbols in the "Word List" column are the standard
description of the NU-6 half-word lists. The number refers to the
50-word list from which the half list is taken. The capital letter
refers to the specific randomization of the order of the 50-word list
(the NU-6 lists are provided in four randomizations). The final,
lowercase letter refers to whether the half list is the first 25 or
the last 25 words in the 50-word list. Thus, list 3Db signifies that
the half list is from the third list of fifty words, in the fourth
(D) randomization of the order, and that it is the last 25 words in
that list. In order to minimize the effects of the word list,
different lists were used for the first nine and the last nine
subjects.

Using this experimental design prevented interaction between the
major effects other than those needed for testing the hypothesis. No
one of the other variables was used with any other more than once per
subject (or order, or list), except for the microphone and noise
level. This means that even if one of the variables had a
significant effect on the raw percent words understood, it would not
have a net effect on the final results.

CHAPTER 4

RESULTS AND ANALYSIS

4.1 General

The results of testing the intelligibility of telephone speech

with the three different microphones at three different background

noise levels are presented in this chapter. The analysis of the

variance shows that the data support the hypothesis that the

proximity effect increases the intelligibility of telephone speech in

noisy environments.

4.2 Pilot Study

The pilot study was conducted in 1984, and provided a

preliminary test of the usefulness of the proximity effect for

increasing telephone intelligibility. The study was not intended to

replicate actual conditions, but merely to give an idea on the

effectiveness of pressure gradient microphones. The telephone

handset and microphones were in a basically free field, and so this

test did not account for the effects of the head on the microphone

response. Three microphones were compared in the pilot. The first

was an omni-directional microphone, which is highly insensitive to

acoustic pressure gradient. The second was a cardioid, which is

somewhat more sensitive to pressure gradient than the pressure

microphone. The last was a bi-directional microphone, which is highly sensitive to the pressure gradient along its major axis. Higher-order gradient microphones are available, but were not used in this investigation.

The results of the pilot, shown in Figure 4.1, are dramatic. The bi-directional microphone maintains very high intelligibility of the telephone speech, even as the noise level is increased from 45 to 65 dBA. The intelligibility with the cardioid microphone degrades somewhat as the noise level is increased. The intelligibility with the omni-directional microphone, however, becomes totally unacceptable at the 65 dBA noise level condition.

While these results were very encouraging, a more realistic experiment which included head effects and a representative microphone position was needed to determine if the proximity effect was of any practical use. The description of the main experiment follows.

4.3 Results

The main experiment was designed to be a more realistic investigation of the use of the proximity effect than the pilot study. The microphones are the same as those used in the pilot study. The main test is designed so that the experimental design is completely blocked for the extraneous effects of subjects, word lists and order. This means that the two major variables, the microphone

Bi-directional

Cardioid

Omni-directional

65

55

45

Background Noise Level (dBA)

100
90
80
70
60
50
40
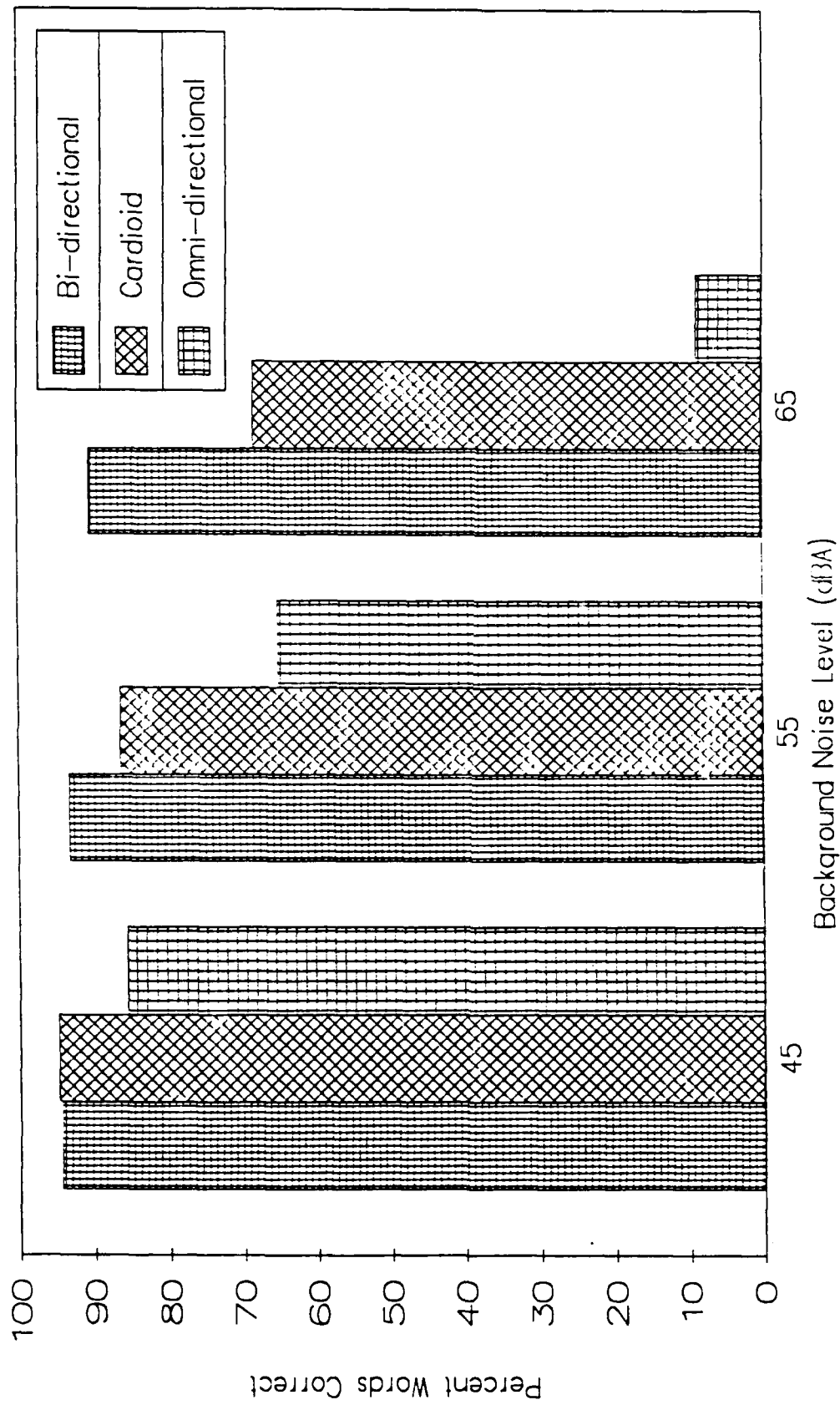30
20
10
0

Percent Words Correct

Figure 4.1 Results of Pilot Study

and the noise level, are not affected by differences between subjects, word lists or order.

After the testing was finished, the words that the subjects had written down on the data collection forms were scored on a correct or incorrect basis. This yielded the percent words understood for each treatment. This percentage was then input to the SAS ANOVA procedure for analysis of variance (SAS User's Guide: Statistics, 1985). The effects due to subject, order, half word list, microphone, noise level, and interaction effects of microphone and noise level were analyzed using the F-test. This test shows the probability that the samples are drawn from the same population (the null hypothesis).

The results of the analysis of variance are shown in Tables 4.1 through 4.7. Table 4.1 shows the results of the F-test and the probability that the null hypothesis is true for the various factors. The results for the order of presentation ($F = 1.05$, $p < 0.4067$) show that the experimental design was effective in minimizing learning effects. The subject, noise level, microphone, word list and the interaction effects of the microphone and the noise level were all significant ($p < 0.0001$).

TABLE 4.1

ANALYSIS OF VARIANCE

Source	: DF :	ANOVA Sum of Squares	: F value	: PR > F:
subject	17	12669	13.54	0.0001
order	8	460	1.05	0.4067
noise level	2	52609	447.80	0.0001
mic	2	10100	91.73	0.0001
list	17	4499	4.81	0.0001
noise*mic	4	2072	9.41	0.0001

TABLE 4.2

SUBJECT EFFECTS (N=9)

SUBJECT	MEAN
1	34.7
2	37.3
3	24.9
4	33.3
5	51.6
6	39.6
7	41.3
8	44.4
9	41.8
10	31.1
11	25.3
12	17.3
13	44.4
14	24.4
15	30.2
16	24.9
17	37.8
18	43.6

TABLE 4.3

EFFECTS OF THE ORDER OF PRESENTATION (N=18)

ORDER	MEAN
1	32.9
2	33.3
3	35.1
4	36.7
5	37.1
6	32.0
7	36.4
8	35.3
9	35.1

TABLE 4.4

EFFECT OF BACKGROUND NOISE LEVEL (N=54)

NOISE LEVEL (dBA)	MEAN
45	56.2
55	36.3
65	12.1

TABLE 4.5

EFFECT OF MICROPHONE TYPE (N=54)

"B" Signifies the Bidirectional Mic;
"C" the Cardoid;
and "O" the Omnidirectional

MIC	MEAN
O	23.8
C	39.5
B	41.4

TABLE 4.6

EFFECT OF WORD LIST USED (N=9)

LIST	MEAN
1Aa	42.7
1Ab	38.7
1Ca	24.9
1Cb	27.1
2Aa	39.6
2Ab	39.1
2Ca	29.8
2Cb	29.3
3Aa	41.8
3Ab	33.3
3Bb	34.7
3Ca	38.7
3Cb	32.4
3Da	34.7
4Aa	40.0
4Ab	39.1
4Ca	34.7
4Cb	27.6

TABLE 4.7

CROSSED EFFECTS OF MICROPHONE AND BACKGROUND NOISE LEVELS
(MEANS) (N=18)

		MICROPHONE		
		OMNI	CARDIOID	BIDIRECTIONAL
NOISE	45:	51.1	59.3	58.2
LEVEL	55:	19.3	42.2	47.3
(dBA)	65:	0.9	16.9	18.7

Since the microphone and noise level have significant F-values
in the analysis of variance, the results were then compared using
Duncan's new multiple range test (Alder and Roessler, 1977). This
method shows that the shortest significant range at the 5% level of

significance is 14.6% words correct. This means that, for example, if there is a 14.6% difference in the scores for two conditions with adjacent means, then there is a 5% probability that the two conditions are drawn from the same population.

The percent words correct for the different conditions are shown in Figure 4.2. The bi-directional and the cardioid microphones were not significantly different, even at the 5% level of significance, at any of the noise levels tested. At the lowest Multi-Talker noise level (45 dBA), the difference between the omni-directional microphone, and the two gradient micropnones is not significant.

At the higher noise levels (55 and 65 dBA) the difference between the gradient microphones and the pressure microphone becomes significant with a 95% confidence interval.

The fact that the subjects and the word lists also have a significant effect on the percent words understood does not bias the data, since the experiment was designed to eliminate such effects. This was done by ensuring that each subject was used for each treatment exactly once. For the same reason, the effect of the word list used does not affect the final analysis.
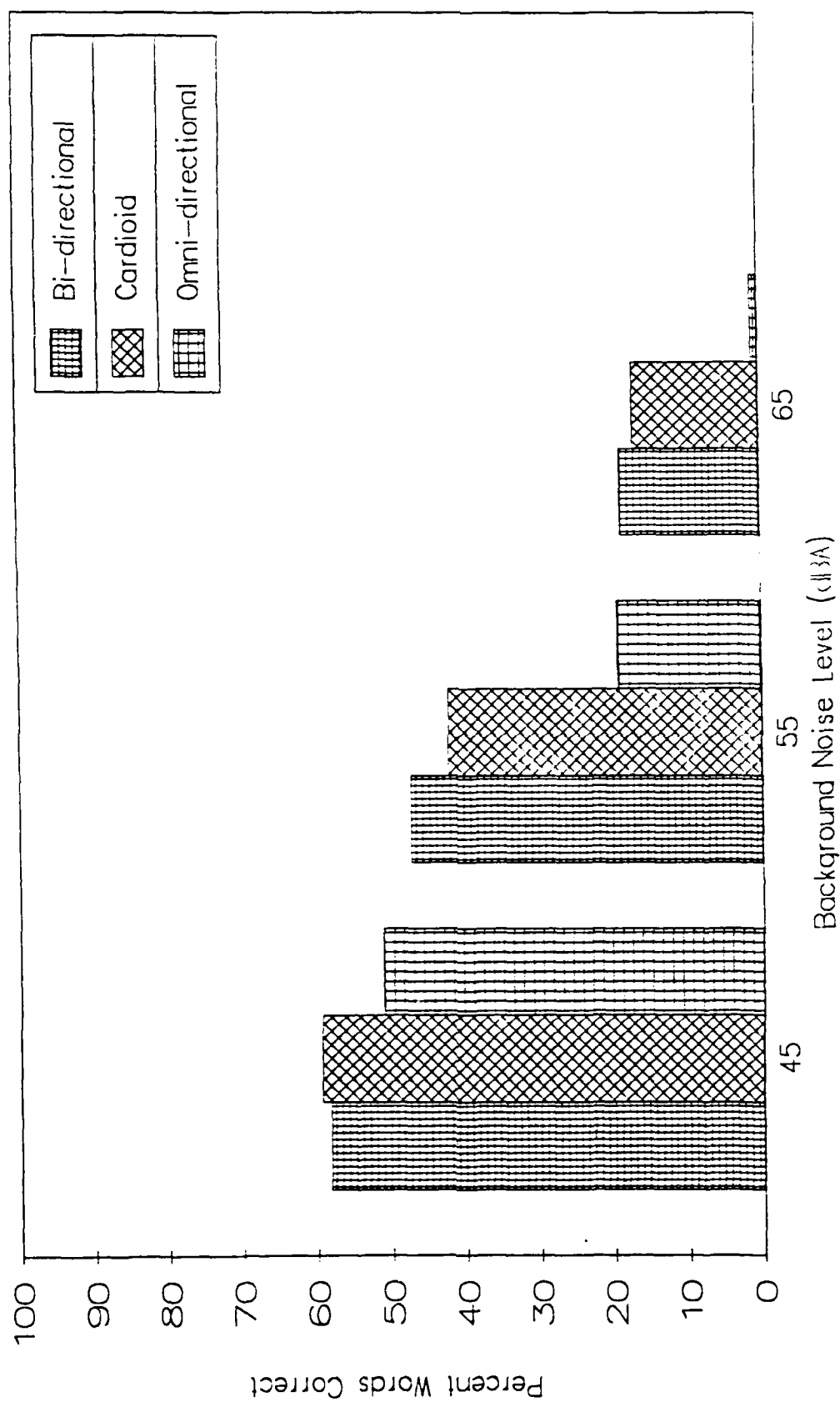
Figure 4.2 Results of Main Experiment

The results of this experiment show that using the proximity effect in hearing aids can increase the intelligibility of telephone speech significantly. At a very loud background noise level (65 dBA), the proximity effect increased the percent words understood from 1% with the omni-directional microphone, to 19% when a gradient microphone is used. Even at the low noise level of 45 dBA, the gradient microphones provided an increase in intelligibility.

4.4 Conclusions and Suggestions For Further Study

The test results from both the pilot study and the main experiment described in this thesis are very promising. In the pilot study, where the microphones were effectively in a free field, the intelligibility was strongly tied to the order of the microphone's sensitivity to the pressure gradient. The bidirectional microphone, which is only a first-order gradient microphone, utilizes the proximity effect to raise the intelligibility of the telephone speech with 65 dBA background noise from totally unintelligible to very intelligible. Even the cardioid microphone, which has gradient sensitivity between first-order and zero-order, maintained the minimum required (70% words understood) for complete understanding of sentences (ANSI S3.14, 1977).

While the results of the main study are not as dramatic as those of the pilot study, they still show that the proximity effect is a

potential tool for dealing with the problem of telephone
communication for the hearing impaired. Even at relatively low noise
levels, the microphones that utilize the proximity effect increase
the intelligibility of the telephone speech. One explanation for the
decrease in the apparent effectiveness of the proximity effect is
that head effects may strongly influence the gradient response of the
microphones. It seems very plausible that the pressure gradient
sensitivity of the microphones is affected by the head at a distance
of 5 mm. or by the telephone 25 mm away. The similarity in the
results of the bi-directional and cardioid microphones indicates that
the improvement due to the proximity effect for this application may
have reached a plateau unless the microphones are designed for
gradient sensitivity when placed near the head. The fact that the
microphones did not provide significantly different scores at the low
(45 dBA) noise level is probably due to experimental error.

This investigation was designed to look at this solution to the
problem of coupling hearing aids and telephones, and shows that the
effect is quite promising. The results show that the use of the
proximity effect in hearing aid microphones provides a significant
improvement in understanding telephone conversation in noise.

Further study into pressure gradient microphones that are
designed for operation in hearing aids, near the head, is needed
before it is known if the proximity effect will be totally successful
in hearing aids. Development of higher-order gradient microphones

may provide even further discrimination against noise, if a method of dealing with the head effects is found. There is also a need for determining the effectiveness of the proximity effect when the hearing aid is used by hearing impaired individuals, rather than by those with normal hearing, and by older subjects as well as young ones.

APPENDIX A

DERIVATION OF THE PROXIMITY EFFECT

The equation for the pressure gradient of a spherical wave can be derived from the equation for pressure. The geometry of the system is shown in Figure A-1. The pressure at a point a distance r from the source is given by

$$p(r) = A \; \frac{\sin(k(ct-r))}{r}$$

where p = pressure

    r = distance from the source to the point

    A = amplitude

    k = wave number = ω/c

    c = the speed of sound

    t = time

    f = frequency

Using a finite difference approximation, the pressure gradient is given by the difference in pressure between two radial points, r_1 and r_2 divided by the spacing (see Figure A-1). The pressure difference is given by:
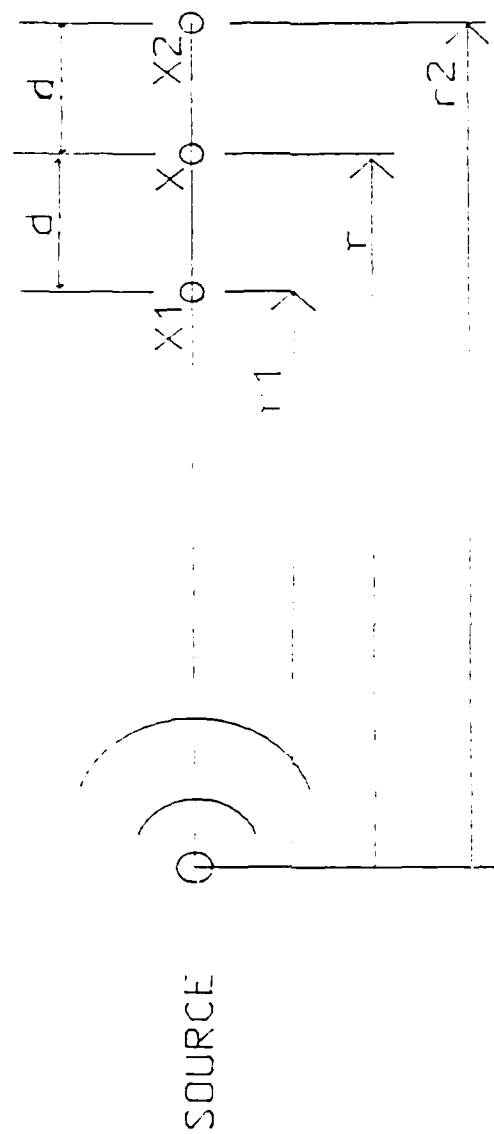
Figure A-1   Geometry of Problem

$$\Delta p = p(r_1) - p(r_2) \qquad \text{where } r_1 = r-d \text{ and } r_2 = r+d$$

$$= \frac{A \sin k(ct-r_1)}{r_1} - \frac{A \sin k(ct-r_2)}{r_2}$$

Rearranging gives:

$$\Delta p = A \left\{ \frac{\sin k(ct-[r-d])}{[r-d]} - \frac{\sin k(ct-[r+d])}{[r+d]} \right\}$$

$$= A \left\{ \frac{[r+d] \sin k(ct-r+d) - [r-d] \sin k (ct-r-d)}{[r-d][r+d]} \right\}$$

$$= \frac{A}{r^2-d^2} \left\{ r[\sin k(ct-r+d)-\sin k(ct-r-d)] \right.$$

$$\left. + d[\sin k(ct-r+d) + \sin k(ct-r-d)] \right\}$$

Using the trigonometric identities,

$$\sin(a+b) - \sin(a-b) = 2\cos(a) \sin(b)$$

and

$$\sin(a+b) + \sin(a-b) = 2 \sin(a) \cos(b)$$

the pressure gradient becomes

$$\frac{\partial p}{\partial r} = \frac{\Delta p}{\Delta r} = \frac{A \{r[2 \cos k(ct-r)\sin kd] + d[2 \sin k(ct-r) \cos kd]\}}{(r^2-d^2)(2d)}$$

The bidirectional microphone used in this experiment has a spacing of 4 mm between faces, so that it is valid to assume that kd << 1 for all frequencies below 5 kHz. This is the frequency range that carries the major portion of telephone and hearing aid speech (Stoker, 1982b; Studebaker and Hochberg, 1980).

The distance from the telephone to the microphone, r_t, is 25 mm;
while the distance from the noise source to the microphone, r_n, is
1.25 n. This means that the term in the denominator (r-d) is much
smaller for the telephone than for the noise, since r_t is of the same
order as d while r_n is much larger than d. This is shown in Figure
A-2, where te difference between the first and second order pressure
gradient and pressure is plotted as a function of kr. The
frequencies associated with values of kr for the telephone (r=25 mm)
and the noise source (r=1.25 m) are shown under the figure. It can
be seen that there is as much as 10 dB increase in ensitivity in the
frequency range of interest.
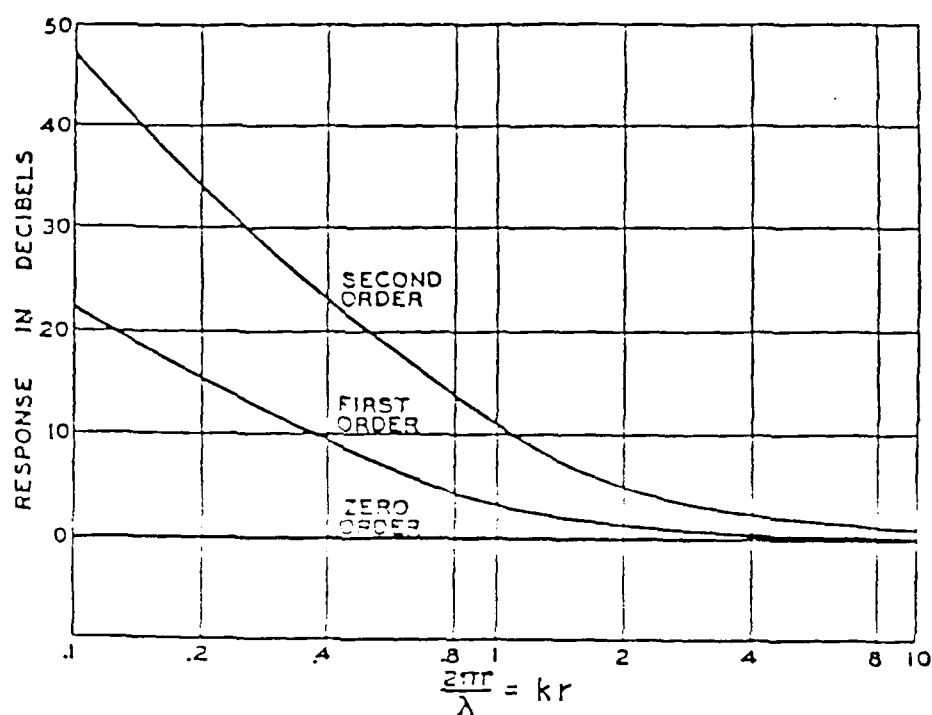
Figure A-2   Response of zero-, first-, and second-order gradient
microphones to a small source as a function of $2\pi r/\lambda$
where r = distance and λ = wavlength. The response
frequency characteristics of all three are assumed to
be independent of the frequency for a plane wave,
that is, $2\pi r/\lambda = \infty$. The frequency scales below the
graph apply to 2 distances, namely 25mm and 1.25m.
(From Olson, 1957)

APPENDIX B

DESCRIPTION OF MULTI-TALKER NOISE

AND INSTRUCTIONS TO SUBJECTS

The multi-talker noise used in this experiment is described in the following paragraph, which forms the appendix to Frank and Craig's "Comparison of the Auditec and Rintelmann Recordings of the NU-6", Journal of Speech and Hearing Disorders, Volume 49. 267-271. August. 1984.

> The multi-talker (MT) noise (available from Auditec of St. Louis) consisted of 20 adult talkers (8 male, 12 female) simultaneously reading different nonemotional magazine articles. The talkers were separated into four groups of five per group. Each group was seated in a separate sound-treated room in a semicircle around a microphone (Shure Model SM-76). The output of each microphone was directed to a four-port mixer (Shure, Professional Mixer) which was directed to a reel-to-reel tape recorder (Crown Model 700). The output of each microphone and Master control was independently adjusted until five listeners all agreed that the speech was unintelligible. The recording was then played back individually to each of the 20 talkers in two 5-minute segments. For each segment each talker rated the intelligibility of the recording on a 5-point scale. A rating of 1 meant that the segment was intelligible in part(s) and a rating of 5 meant that the entire segment was unintelligible. The mean ratings were 4.90 and 4.94 for the two segments respectively. The MT noise recording was then played (Crown Model 700) to a spectrometer (Bruel & Kjaer Model 2111)

which was connected to a level recorder (Bruel &
Kjaer Model 2305) set at a lower limiting
frequency of 20 Hz, having a writing speed of 315
mm/s, paper speed of 10 mm/s, and potentiometer
range of 50 dB. The MT noise recording was then
played and recorded on the level recorder while
the spetrometer was set on the linear
response. The local maxima of the root-mean-
square (rms) values were very stable ($\pm$ 2.5 dB)
for the entire recording. A 1000-Hz calibration
tone was then recorded at the average local
maxima rms value and spliced onto the beginning
of the recording. The MT noise recording was
then played through the spectrometer and level
recorder so that a 1/3 octave-band analysis could
be completed from 100 to 5000 Hz. The average
local maxima rms values within each 1/3 octave
band were then determined and the spectrum was
found to be flat ($\pm$ 2.5 dB) from 100 to 1000 Hz
and decreasing thereafter about 6 dB-octave.
Following this the recording was copied 10 times
and spliced together to produce a continuous 60-
minute tape. The spliced copy was then recopied
to form the MT noise used in this study.

<u>INSTRUCTIONS TO SUBJECTS</u>

Use of the Proximity Effect to Improve Hearing Aid User's Ability to Use Telephones

Investigators: C. P. Janota, Research Associate
Applied Research Laboratory

K. R. Wilson, Graduate Assistant
Acoustics Program, Applied Research Laboratory

We are attempting to improve the intelligibility of telephone speech for hearing aid users in many environments by increasing the signal-to-noise ratio. We hope to show that this can be done by utilizing the proximity effect at the hearing aid microphone. You will be asked to record word lists that you hear through the telephone hearing aid system with various combinations of microphones and background noise level. Please familiarize yourself with the attached list of words prior to your first session. The word lists say "<u>say</u> the word _____." but you will be asked to write the word on an answer sheet. If you are unsure of the word or spelling, guess! Your responses will be tallied to give the intelligibility of each microphone-noise level combination. Complete anonymity will be maintained; and the final results will be the average of all subjects.

We ask you to schedule three testing sessions, each taking around 75 minutes and comprising nine conditions. The tests will take place in the sound proof chamber in 119 ARL. A modified hearing aid will be inserted in one ear. The insertion and removal of the aid are mildly uncomfortable, but once it is in, it is barely noticeable. Try to concentrate on the tests, even though they are monotonous and the noise is distracting. If you wish to end a session for any reason, remove the aid or signal me through the window.

Please fill out the attached informed consent forms - one copy is for you, and the other is permanently stored by the Office of the Vice-President for Research and Graduate Studies. If you have any problems, questions or concerns, contact me.

K. Wilson   119 ARL   3-3214

Figure B-1. Pilot Study: Instructions to Subjects

INSTRUCTIONS TO SUBJECTS

Exploitation of the Proximity Effect to Improve Hearing

Aid User's Ability to Hear Telephone Conversation

Investigators: C. P. Janota, Research Associate
Applied Research Laboratory

K. R. Wilson, Graduate Assistant
Acoustics Program, Applied Research Lab.

We are attempting to improve the intelligibility of telephone speech for hearing aid users in noisy rooms by increasing the signal-to-noise ratio. We hope to show that this can be done by using a mic that is more sensitive to nearby sounds than the noise. You will be asked to record word lists that you hear through the telephone-hearing aid system. The number of words that you understand is a measure of the intelligibility of the microphones.

At the beginning of the test, I will insert a modified hearing-aid into your left ear. You will then hear a man reading a short story, and some background noise. His speech will gradually become louder. I want you to tell me when the speech is at the most comfortable level for you, that is, when the speech is at the level that you most prefer for listening.

The test will consist of 9 lists, with different mics and noise levels. Each list has 25 words. The lists say, "say the word ___," but I want you to write the word down on the answer sheet. If you are unsure of a word or its spelling, guess! If you have no idea, put a line through the number so that you do not lose your place. Sometimes it will be difficult, if not impossible. Don't worry - it is meant to be that way. No one understands much with some mics. Other times it will be very easy. Remember, I am not testing you, WE are testing the mics. Try to concentrate on the tests, even though it is monotonous, and the noise is distracting. If you wish to end a session, signal me through the window.

Please fill out the attached informed consent forms - one copy is for you and the other is permanently stored by the Office of the Vice-President for Research and Graduate Studies.

Any questions?

Kathryn Wilson   121 ARL   865-6367

Figure B-2. Instructions to Subjects

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